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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/829,314	04/09/2001	Steven C. Dzik	Dzik 7	7112

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EXAMINER	
PHILPOTT, JUSTIN M	

ART UNIT	PAPER NUMBER
2665	

DATE MAILED: 11/22/2005

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	09/829,314	DZIK, STEVEN C.
	Examiner	Art Unit
	Justin M. Philpott	2665

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *18 October 2005*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-35* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-35* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTION

Continued Examination Under 37 CFR 1.114

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on October 18, 2005 has been entered.

Response to Arguments

2. Applicant's arguments with respect to claims 1-35 have been considered but are moot in view of the new ground(s) of rejection. Specifically, applicant's arguments that the new limitations added to the amended claims are not taught by the previously cited art are moot since these new limitations are taught by the newly cited art as discussed in the following office action.

Claim Rejections - 35 USC § 103

3. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. Claims 1-35 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent Application Publication No. US 2003/0112796 by Kwan in view of U.S. Patent No. 6,356,545 to Vargo et al.

5. Regarding claim 1, Kwan teaches a method of processing a sequence of audio samples, each of the samples being stored within a respective packet, the method comprising: retrieving a first packet from an input buffer (e.g., see paragraph 0227 regarding generating voice parameters based upon buffered voice samples), the first packet implicitly having an associated length; determining pitch associated with audio information contained within the first packet (e.g., see paragraph 0230 regarding calculating the pitch associated with the voice sample); determining whether a second packet of the audio information has arrived at the input buffer (e.g., see paragraph 0228 regarding experiencing a loss of a subsequent packet containing voice samples), the second packet implicitly having an expected arrival time (e.g., see paragraph 0224 regarding packet arriving too late); and adjusting the first packet using at least the pitch (e.g., see paragraphs 0243-0252 regarding pitch period and voicing calculation and paragraphs 0223-0242 regarding determining when the subsequent packet is lost, or not timely arrived).

However, Kwan may not specifically disclose adjusting the length of a first packet using at least one pitch period associated with the pitch of the first packet upon determining that a second packet arrives after an expected arrival time.

Vargo, like Kwan, teaches a method of processing audio samples (e.g., see abstract), and further, teaches adjusting the length of a packet (e.g., stretching the data remaining in the buffer, see col. 11, lines 34-47) using at least one pitch period associated with the pitch of the previous packet (e.g., see col. 11, lines 47-52 regarding stretching without changing pitch with respect to

the previous packet) upon determining that a next packet arrives after an expected arrival time (e.g., see col. 11, lines 33-52 regarding determining that a subsequent packet for arrival at the buffer has not arrived in time which will result in a gap in transmission if not compensated for). The teachings of Vargo provide improved speech quality and maintaining consistent pitch and delay-free voice transmission (e.g., see col. 2, lines 49-63). Thus, at the time of the invention it would have been obvious to one of ordinary skill in the art to apply the audio processing teachings of Vargo to the audio processing method of Kwan in order to provide improved speech quality and maintaining consistent pitch and delay-free voice transmission.

Regarding claim 2, Kwan teaches adjusting comprises processing at least two pitch periods to produce a new pitch period (e.g., see paragraphs 0243-0252 regarding calculating pitch periods over a range of pitch values).

Regarding claim 3, Kwan teaches the new pitch period replaces the at least two adjacent periods (e.g., see paragraphs 0267-0268 wherein replacement may occur for more than one lost packet).

Regarding claim 4, Kwan teaches the new pitch period is inserted between two of at least two adjacent periods (e.g., see paragraph 0268).

Regarding claim 5, Kwan teaches determining a scheduled play out time of the audio information within the second packet (e.g., see paragraph 0218 and 0222 regarding determining target hold times and voice synchronizer).

Regarding claim 6, Kwan teaches determining an estimated time of arrival of a sequentially following packet (e.g., see paragraph 0217 regarding voice traffic comprising isochronous transmission).

Regarding claim 7, Kwan teaches a target play time comprises the ETA and a latency period of the sequentially following packet (e.g., see paragraphs 0217-0218 regarding target hold times and isochronous transmission).

Regarding claim 8, Kwan teaches the play out time of audio information within the second packet is reduced in response to an early arrival of a sequentially following packet at the input buffer (e.g., see paragraph 0220 regarding decreasing the holding time).

Regarding claim 9, Kwan teaches the play out time of audio information within the second packet is not reduced by a factor greater than two (e.g., see paragraph 0220 regarding decreasing the holding time by transferring only one of the voice frames to the media queue).

Regarding claim 10, Kwan teaches the play out time of audio information within the second packet is reduced by deleting at least one pitch period of a plurality of pitch periods contained within the audio information (e.g., see paragraph 0244-0253, 0258 and 0268 regarding stretching pitch periods to cover gaps in time due to lost packets, whereby a pitch period is deleted).

Regarding claim 11, Kwan teaches the target play time of audio information within the second packet is expanded if a next packet arrives during a latency period associated with the next packet (e.g., see paragraph 0228 regarding elapsing of a timeout period).

Regarding claim 12, Kwan teaches a play time of audio information within the second packet is adjusted to compensate for adjustments of play time of the retrieved packet (e.g., see paragraph 0228, 0244-0252, 0258 and 0268 regarding determining a pitch period, and synthesizing voice based on the pitch period).

Regarding claim 13, Kwan teaches an apparatus comprising: a first VoIP gateway (e.g., see paragraph 0077 regarding gateway and the system comprising VoIP) for retrieving a first packet from an input buffer, the packet implicitly having an associated length; the first VoIP gateway (e.g., see paragraph 0077) determining pitch associated with audio information contained within the first packet (e.g., see paragraph 0230 regarding calculating the pitch associated with the voice sample); the first VOIP gateway determining whether a second packet of the audio information has arrived at the input buffer (e.g., see paragraph 0228 regarding experiencing a loss of a subsequent packet containing voice samples), the second packet implicitly having an expected arrival time (e.g., see paragraph 0224 regarding packet arriving too late); and adjusting the first packet using at least the pitch (e.g., see paragraphs 0243-0252 regarding pitch period and voicing calculation and paragraphs 0223-0242 regarding determining when the subsequent packet is lost, or not timely arrived).

However, as discussed above regarding claim 1, Kwan may not specifically disclose adjusting the length of a first packet using at least one pitch period associated with the pitch of the first packet upon determining that a second packet arrives after an expected arrival time.

Vargo, like Kwan, teaches a method of processing audio samples (e.g., see abstract), and further, teaches adjusting the length of a packet (e.g., stretching the data remaining in the buffer, see col. 11, lines 34-47) using at least one pitch period associated with the pitch of the previous packet (e.g., see col. 11, lines 47-52 regarding stretching without changing pitch with respect to the previous packet) upon determining that a next packet arrives after an expected arrival time (e.g., see col. 11, lines 33-52 regarding determining that a subsequent packet for arrival at the buffer has not arrived in time which will result in a gap in transmission if not compensated for).

The teachings of Vargo provide improved speech quality and maintaining consistent pitch and delay-free voice transmission (e.g., see col. 2, lines 49-63). Thus, at the time of the invention it would have been obvious to one of ordinary skill in the art to apply the audio processing teachings of Vargo to the audio processing method of Kwan in order to provide improved speech quality and maintaining consistent pitch and delay-free voice transmission.

Regarding claim 14, Kwan teaches the adjusting comprises processing at least two adjacent pitch periods to produce a new pitch period (e.g., see paragraphs 0243-0252 regarding calculating pitch periods over a range of pitch values).

Regarding claim 15, Kwan teaches the new pitch period replaces the at least two adjacent pitch periods (e.g., see paragraphs 0267-0268 wherein replacement may occur for more than one lost packet).

Regarding claim 16, Kwan teaches the new pitch period is inserted between two of at least two adjacent periods (e.g., see paragraph 0268).

Regarding claim 17, Kwan teaches the gateway determines a scheduled play out time of the audio information within the second packet (e.g., see paragraph 0218 and 0222 regarding determining target hold times and voice synchronizer).

Regarding claim 18, Kwan teaches the gateway determines an estimated time of arrival of a sequentially following packet (e.g., see paragraph 0217 regarding voice traffic comprising isochronous transmission).

Regarding claim 19, Kwan teaches a target play time comprises the ETA and a latency period of the sequentially following packet (e.g., see paragraphs 0217-0218 regarding target hold times and isochronous transmission).

Regarding claim 20, Kwan teaches the scheduled play out time of audio information within the second packet is reduced in response to an early arrival of a sequentially following packet at the input buffer (e.g., see paragraph 0220 regarding decreasing the holding time).

Regarding claim 21, Kwan teaches the scheduled play out time of audio information within the second packet is not reduced by a factor greater than two (e.g., see paragraph 0220 regarding decreasing the holding time by transferring only one of the voice frames to the media queue).

Regarding claim 22, Kwan teaches the scheduled play out time of audio information within the second packet is reduced by deleting at least one pitch period of a plurality of pitch periods contained within the audio information (e.g., see paragraph 0244-0253, 0258 and 0268 regarding stretching pitch periods to cover gaps in time due to lost packets, whereby a pitch period is deleted).

Regarding claim 23, Kwan teaches the target play time of audio information within the second packet is expanded if a next packet arrives during the latency period of the sequentially following packet (e.g., see paragraph 0228 regarding elapsing of a timeout period).

Regarding claim 24, Kwan teaches the target play time of audio information within the second packet is expanded by copying pitch periods contained within the audio information of the second packet (e.g., see paragraph 0228, 0244-0252, 0258 and 0268 regarding determining a pitch period, and synthesizing voice based on the pitch period).

Regarding claim 25, Kwan teaches an apparatus for expanding and reducing audio information within packets, comprising: a processor (e.g., selector 196 within lost packet recovery engine, see paragraph 0226 and FIG. 12); and a storage device (e.g., voice analyzer

192, see paragraph 0227) coupled (e.g., via 194) to the processor (e.g., selector 196 within lost packet recovery engine, see paragraph 0226 and FIG. 12) for controlling the processor, the processor comprising instructions operative to: retrieve a first packet from an input buffer (e.g., see paragraph 0227 regarding generating voice parameters based upon buffered voice samples), the first packet implicitly having an associated length; determine pitch associated with audio information contained within the first packet (e.g., see paragraph 0230 regarding calculating the pitch associated with the voice sample); determine whether a second packet of the audio information has arrived at the input buffer (e.g., see paragraph 0228 regarding experiencing a loss of a subsequent packet containing voice samples), the second packet implicitly having an expected arrival time (e.g., see paragraph 0224 regarding packet arriving too late); and adjusting the first packet using at least the pitch (e.g., see paragraphs 0243-0252 regarding pitch period and voicing calculation and paragraphs 0223-0242 regarding determining when the subsequent packet is lost, or not timely arrived).

However, as discussed above regarding claim 1, Kwan may not specifically disclose adjusting the length of a first packet using at least one pitch period associated with the pitch of the first packet upon determining that a second packet arrives after an expected arrival time.

Vargo, like Kwan, teaches a method of processing audio samples (e.g., see abstract), and further, teaches adjusting the length of a packet (e.g., stretching the data remaining in the buffer, see col. 11, lines 34-47) using at least one pitch period associated with the pitch of the previous packet (e.g., see col. 11, lines 47-52 regarding stretching without changing pitch with respect to the previous packet) upon determining that a next packet arrives after an expected arrival time (e.g., see col. 11, lines 33-52 regarding determining that a subsequent packet for arrival at the

buffer has not arrived in time which will result in a gap in transmission if not compensated for). The teachings of Vargo provide improved speech quality and maintaining consistent pitch and delay-free voice transmission (e.g., see col. 2, lines 49-63). Thus, at the time of the invention it would have been obvious to one of ordinary skill in the art to apply the audio processing teachings of Vargo to the audio processing method of Kwan in order to provide improved speech quality and maintaining consistent pitch and delay-free voice transmission.

Regarding claim 26, Kwan teaches computer readable medium having stored thereon a plurality of instructions including instructions which, when executed by a processor (e.g., selector 196 within lost packet recovery engine, see paragraph 0226 and FIG. 12), ensures the processor to perform a method comprising: retrieving a first packet from an input buffer (e.g., see paragraph 0227 regarding generating voice parameters based upon buffered voice samples), the first packet implicitly having an associated length; determining pitch associated with audio information contained within the packet (e.g., see paragraph 0230 regarding calculating the pitch associated with the voice sample), the second packet implicitly having an expected arrival time (e.g., see paragraph 0224 regarding packet arriving too late); and adjusting the first packet using at least the pitch (e.g., see paragraphs 0243-0252 regarding pitch period and voicing calculation and paragraphs 0223-0242 regarding determining when the subsequent packet is lost, or not timely arrived).

However, as discussed above regarding claim 1, Kwan may not specifically disclose adjusting the length of a first packet using at least one pitch period associated with the pitch of the first packet upon determining that a second packet arrives after an expected arrival time.

Vargo, like Kwan, teaches a method of processing audio samples (e.g., see abstract), and further, teaches adjusting the length of a packet (e.g., stretching the data remaining in the buffer, see col. 11, lines 34-47) using at least one pitch period associated with the pitch of the previous packet (e.g., see col. 11, lines 47-52 regarding stretching without changing pitch with respect to the previous packet) upon determining that a next packet arrives after an expected arrival time (e.g., see col. 11, lines 33-52 regarding determining that a subsequent packet for arrival at the buffer has not arrived in time which will result in a gap in transmission if not compensated for). The teachings of Vargo provide improved speech quality and maintaining consistent pitch and delay-free voice transmission (e.g., see col. 2, lines 49-63). Thus, at the time of the invention it would have been obvious to one of ordinary skill in the art to apply the audio processing teachings of Vargo to the audio processing method of Kwan in order to provide improved speech quality and maintaining consistent pitch and delay-free voice transmission.

Regarding claim 27, Kwan discloses retrieving a first packet from an input buffer (e.g., see paragraph 0227 regarding generating voice parameters based upon buffered voice samples); determining a pitch within audio samples for the retrieved packet (e.g., see paragraph 0230 regarding calculating the pitch associated with the voice sample); and adjusting the first packet using at least the pitch (e.g., see paragraphs 0243-0252 regarding pitch period and voicing calculation and paragraphs 0223-0242 regarding determining when the subsequent packet is lost, or not timely arrived).

However, as discussed above regarding claim 1, Kwan may not specifically disclose adjusting the length of a first packet using at least one pitch period associated with the pitch of the first packet upon determining that a second packet arrives after an expected arrival time.

Vargo, like Kwan, teaches a method of processing audio samples (e.g., see abstract), and further, teaches adjusting the length of a packet (e.g., stretching the data remaining in the buffer, see col. 11, lines 34-47) using at least one pitch period associated with the pitch of the previous packet (e.g., see col. 11, lines 47-52 regarding stretching without changing pitch with respect to the previous packet) upon determining that a next packet arrives after an expected arrival time (e.g., see col. 11, lines 33-52 regarding determining that a subsequent packet for arrival at the buffer has not arrived in time which will result in a gap in transmission if not compensated for). The teachings of Vargo provide improved speech quality and maintaining consistent pitch and delay-free voice transmission (e.g., see col. 2, lines 49-63). Thus, at the time of the invention it would have been obvious to one of ordinary skill in the art to apply the audio processing teachings of Vargo to the audio processing method of Kwan in order to provide improved speech quality and maintaining consistent pitch and delay-free voice transmission.

Regarding claims 28-30, Kwan discloses that the voice traffic is sent from a far end in an isochronous manner, meaning one packet after the other without delay (e.g., see paragraph 0217). The estimated time of arrival of a sequentially following packet is necessarily immediately after a currently received packet. Thus, the target holding time of Kwan comprises an estimated arrival time as well as an estimated worst case jitter, which meets the limitation of a latency.

Regarding claims 31 and 32, Kwan discloses expanding a play time of a received packet when the sequentially following packet arrives during its latency period by synthesizing voice until the voice decoder receives a voice packet, or a timeout period has elapsed (e.g., see

paragraph 0228). This synthesizing of voice requires determining a pitch period and synthesizing voice based on the pitch period (e.g., see paragraphs 0244-0252, 0258, and 0268).

Regarding claim 33, Kwan discloses decreasing the holding time rapidly to minimize excessive end to end delay, which is accomplished by passing two voice frames to the voice decoder in one decoding interval but only one of the voice frames is transferred to the media queue (e.g., see paragraph 0220). This meets the limitation of reducing the play time of a packet. Kwan does not disclose reducing the play time by greater than a factor of two.

Regarding claim 34, Kwan discloses that two voice frames may be sent to the voice decoder, and only one may be sent to the media queue in order to compress the voice data, as mentioned above. Also, Kwan discloses stretching pitch periods to cover gaps in time due to lost packets (e.g., see paragraphs 0244-0252, 0258, and 0268). It follows that when only one frame is played, when normally two would be played, that one frame is deleted, thus a pitch period is deleted.

Regarding claim 35, Kwan discloses that the reducing of holding times may be performed in response to excessive end to end delays created by long holding times to compensate for the step of expanding (e.g., paragraph 0220).

Conclusion

6. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. U.S. Patent Nos. 5,699,485 to Shoham, 5,825,771 to Cohen et al. and 5,890,108 to Yeldener each disclose audio processing methods with adjusting pitch periods.

7. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Justin M. Philpott whose telephone number is 571.272.3162. The examiner can normally be reached on M-F, 9:00am-5:00pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Huy D. Vu can be reached on 571.272.3155. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

Justin M Philpott

ALPUS H. HSU
PRIMARY EXAMINER